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H4K KTKX

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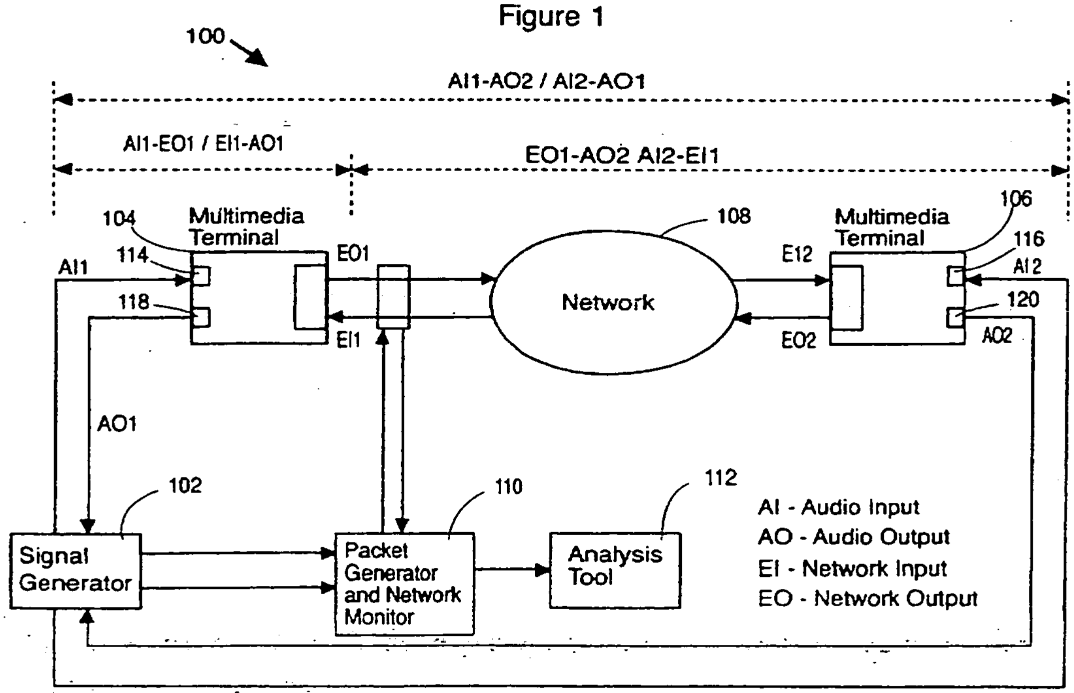
(54) Abstract Title
Measuring the delay of audio stream packets across a network using marker packets

(57) A system is provided for measuring delays across equipment in a packet network. A test audio signal is generated by a signal generator 102. A multimedia terminal 104 receives and digitally encodes the test audio signal forming an audio stream packet for transmission onto the network 108. Multimedia terminal 106 receives and decodes the audio stream packet and regenerates test audio signal which is then looped back to the signal generator 102. At various stages (e.g. test signal generation, audio stream packet transmission), marker packets are generated by the packet generator 110 in response to trigger signals from the signal generator 102. Marker packets and audio stream packets are time-stamped and logged for analysis. Time differences between marker and audio stream packets determine delays at various stages. Measurement accuracy can be improved by locating the position of the encoded test signal in the audio stream packets.

Figure 1



At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

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Figure 1

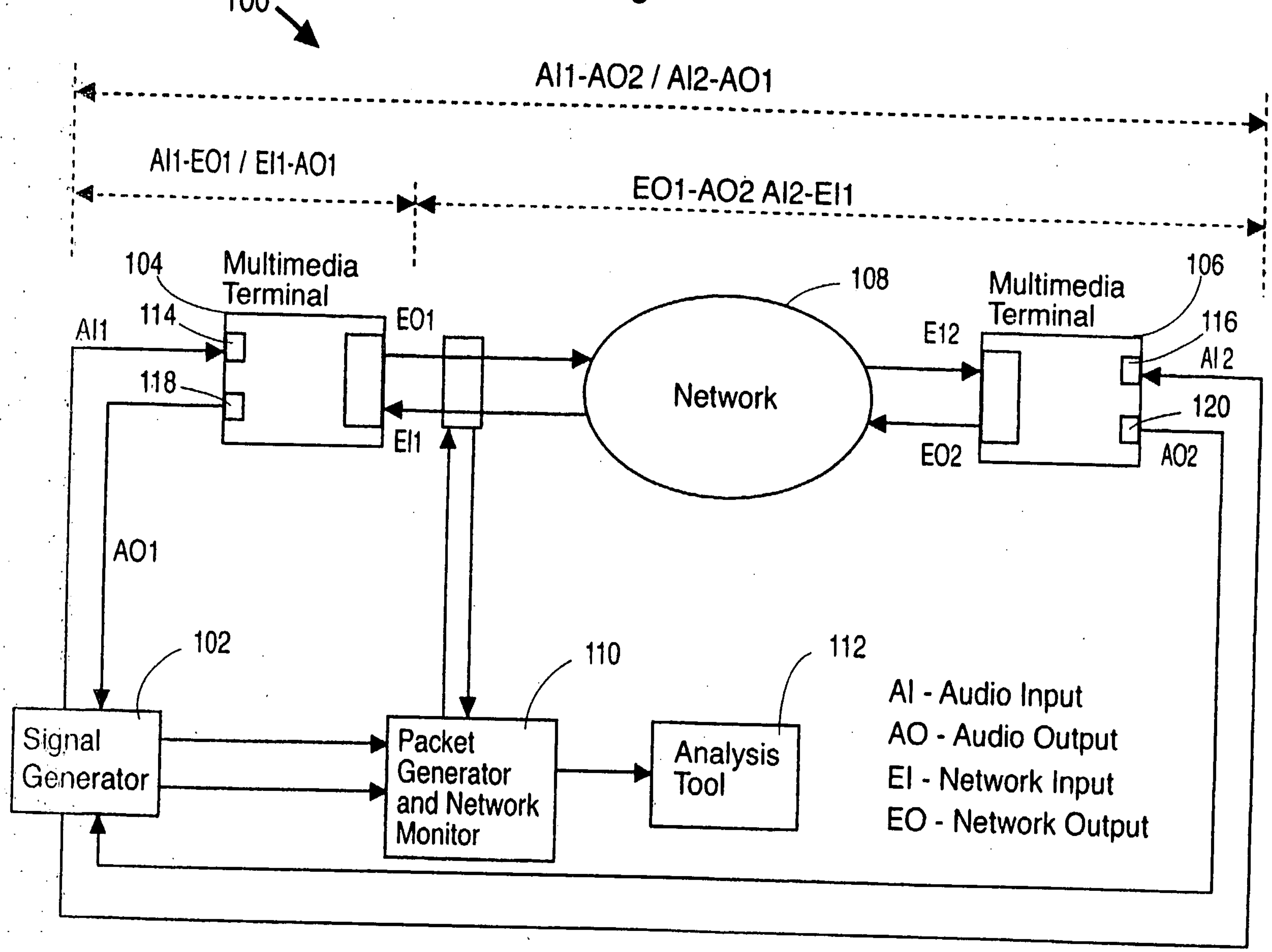

100
AI1-AO2 / AI2-AO1
AI1-EO1 / EI1-AO1
EO1-AO2 AI2-EI1
104
Multimedia Terminal
114
118
AI1
EO1
EI1
108
Network
106
Multimedia Terminal
116
AI 2
120
AO2
E12
EO2
AO1
102
Signal Generator
110
Packet Generator and Network Monitor
112
Analysis Tool
AI - Audio Input
AO - Audio Output
EI - Network Input
EO - Network Output

Figure 2

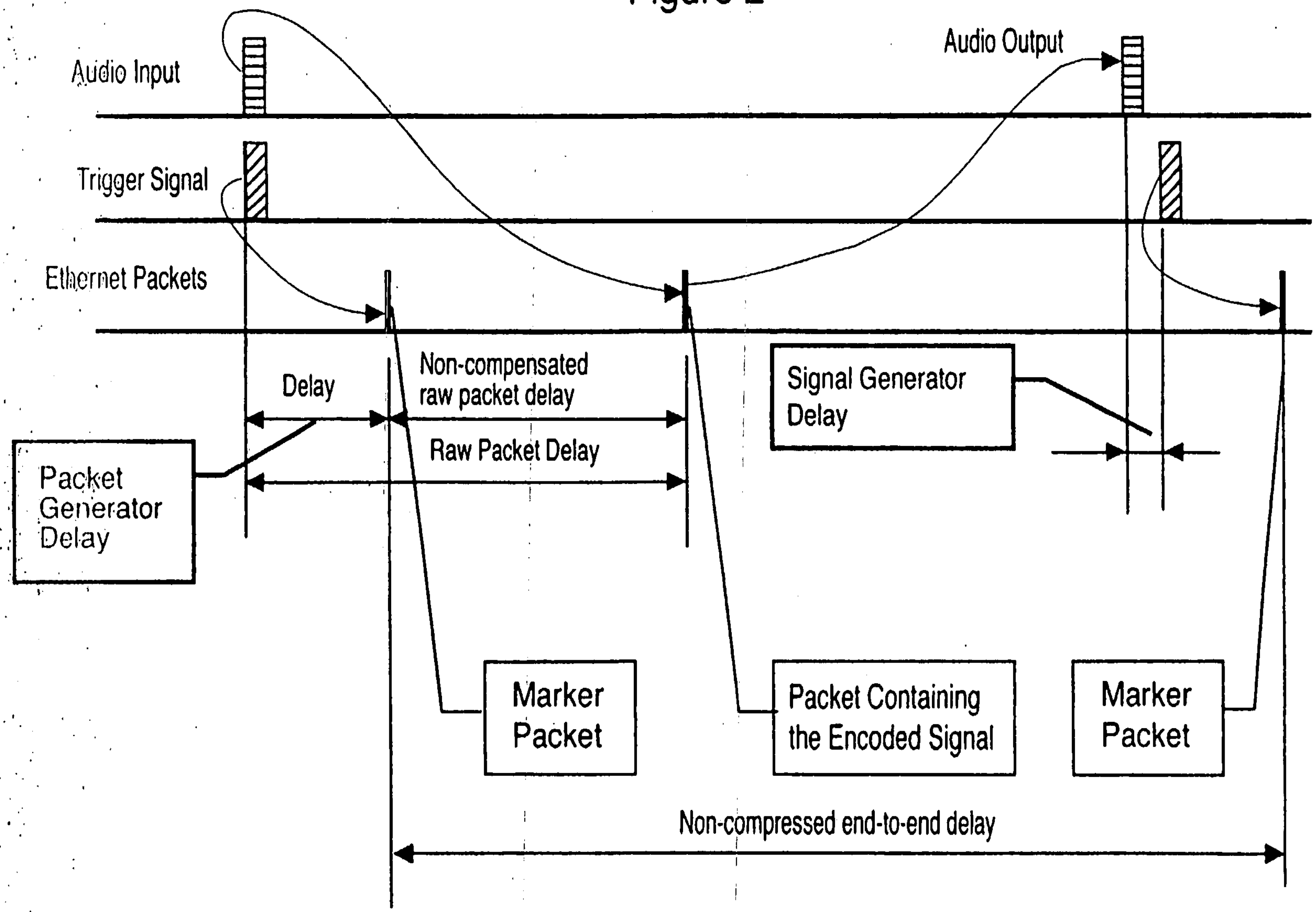

Audio Input
Trigger Signal
Ethernet Packets
Audio Output
Delay
Non-compensated raw packet delay
Raw Packet Delay
Packet Generator Delay
Signal Generator Delay
Marker Packet
Packet Containing the Encoded Signal
Marker Packet
Non-compressed end-to-end delay

Figure 3A

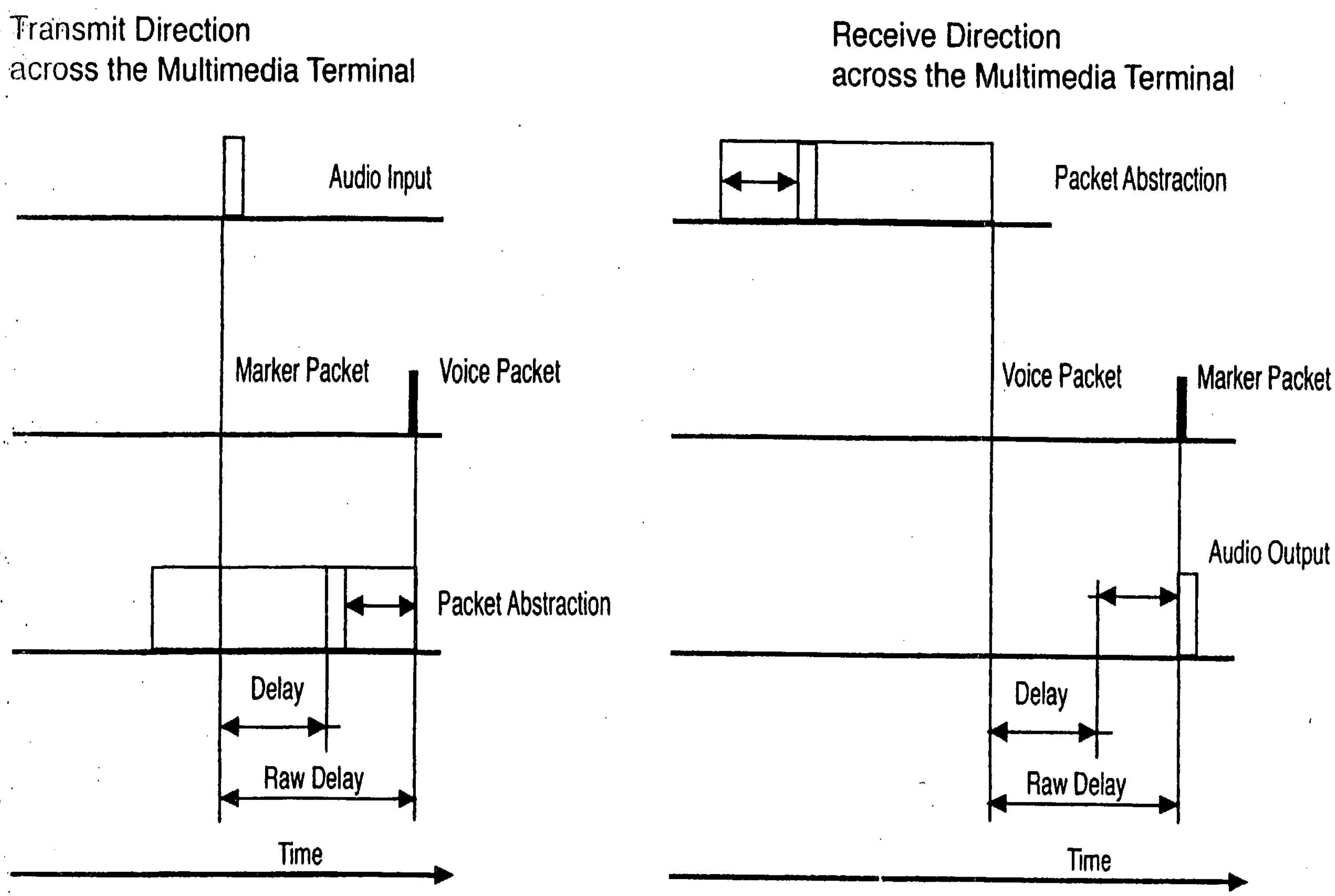

Transmit Direction
across the Multimedia Terminal
Audio Input
Marker Packet
Voice Packet
Packet Abstraction
Delay
Raw Delay
Time
Receive Direction
across the Multimedia Terminal
Packet Abstraction
Voice Packet
Marker Packet
Audio Output
Delay
Raw Delay
Time

Figure 3B - No Delays

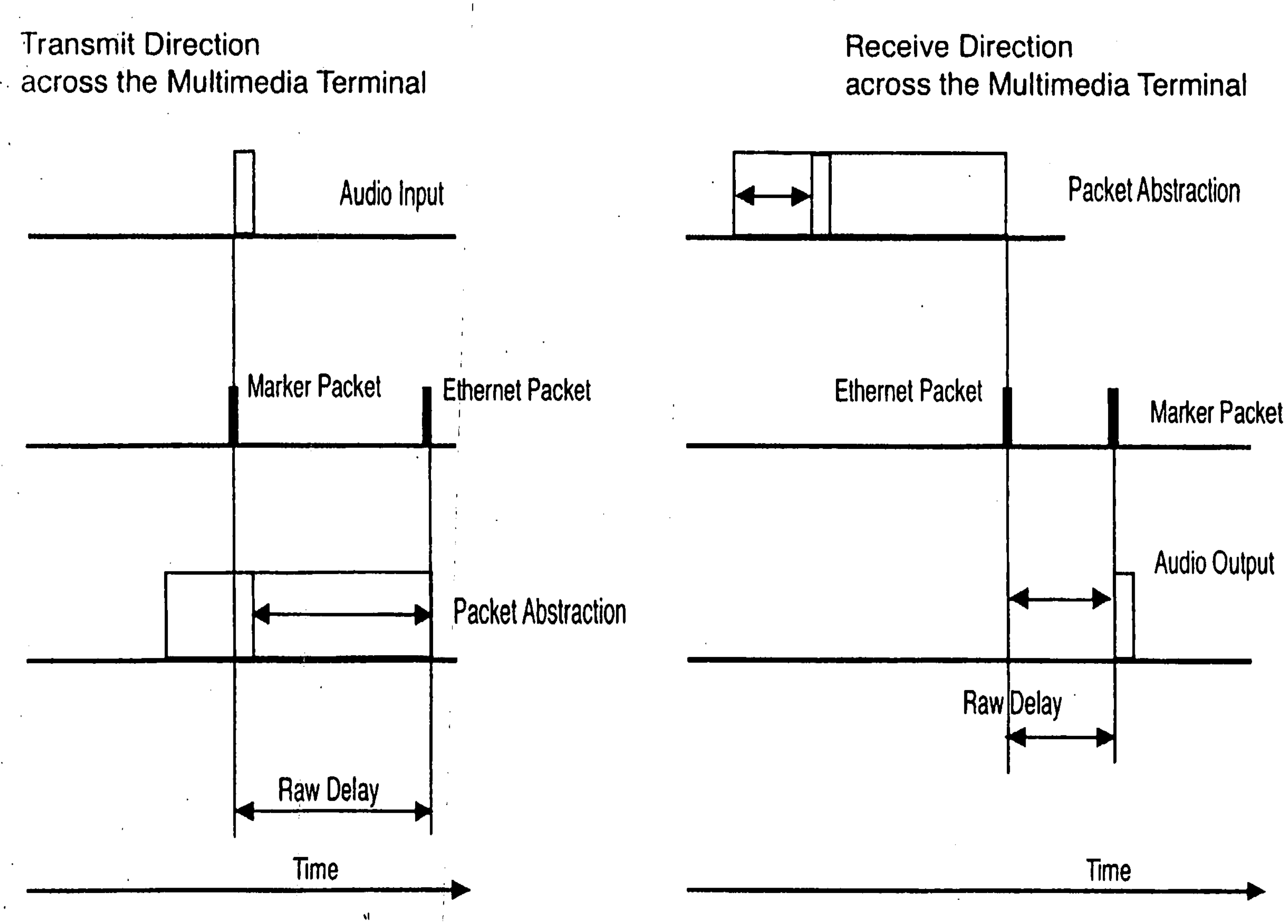


Figure 4

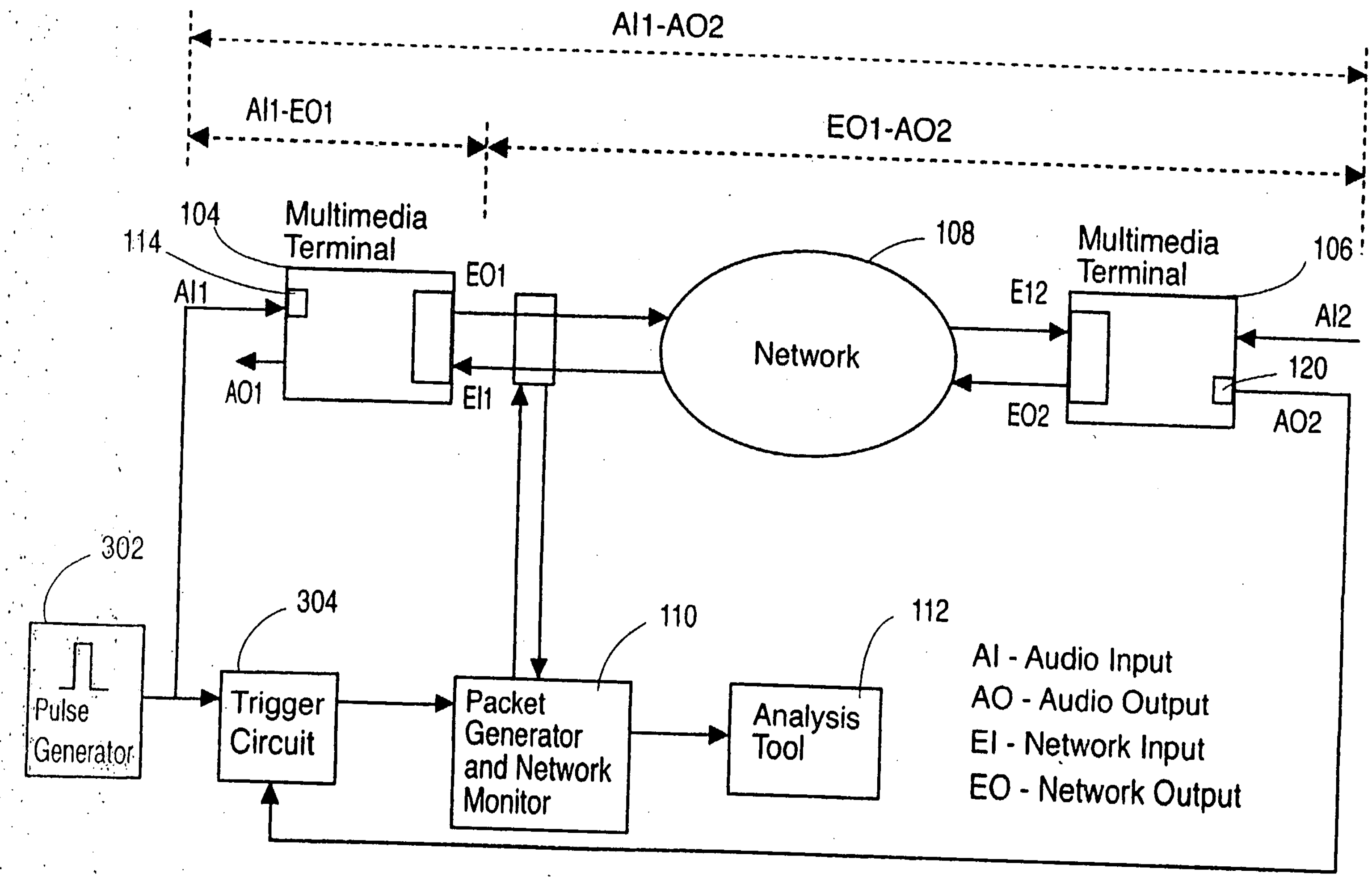

AI1-AO2
AI1-EO1
EO1-AO2
104
Multimedia Terminal
114
AI1
AO1
EO1
EI1
108
Network
Multimedia Terminal
106
E12
EO2
AI2
120
AO2
302
Pulse Generator
304
Trigger Circuit
110
Packet Generator and Network Monitor
112
Analysis Tool
AI - Audio Input
AO - Audio Output
EI - Network Input
EO - Network Output

Figure 4A

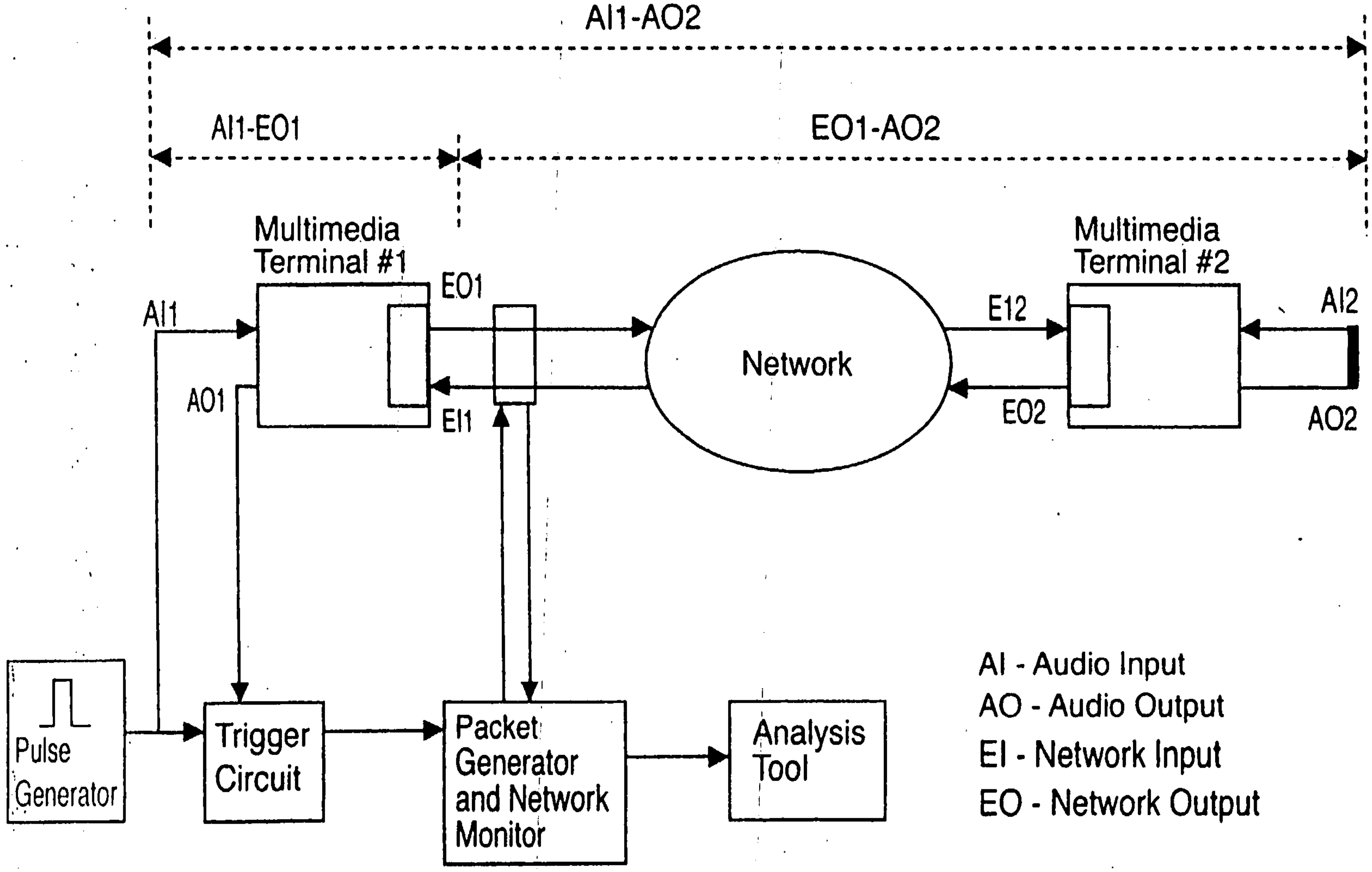
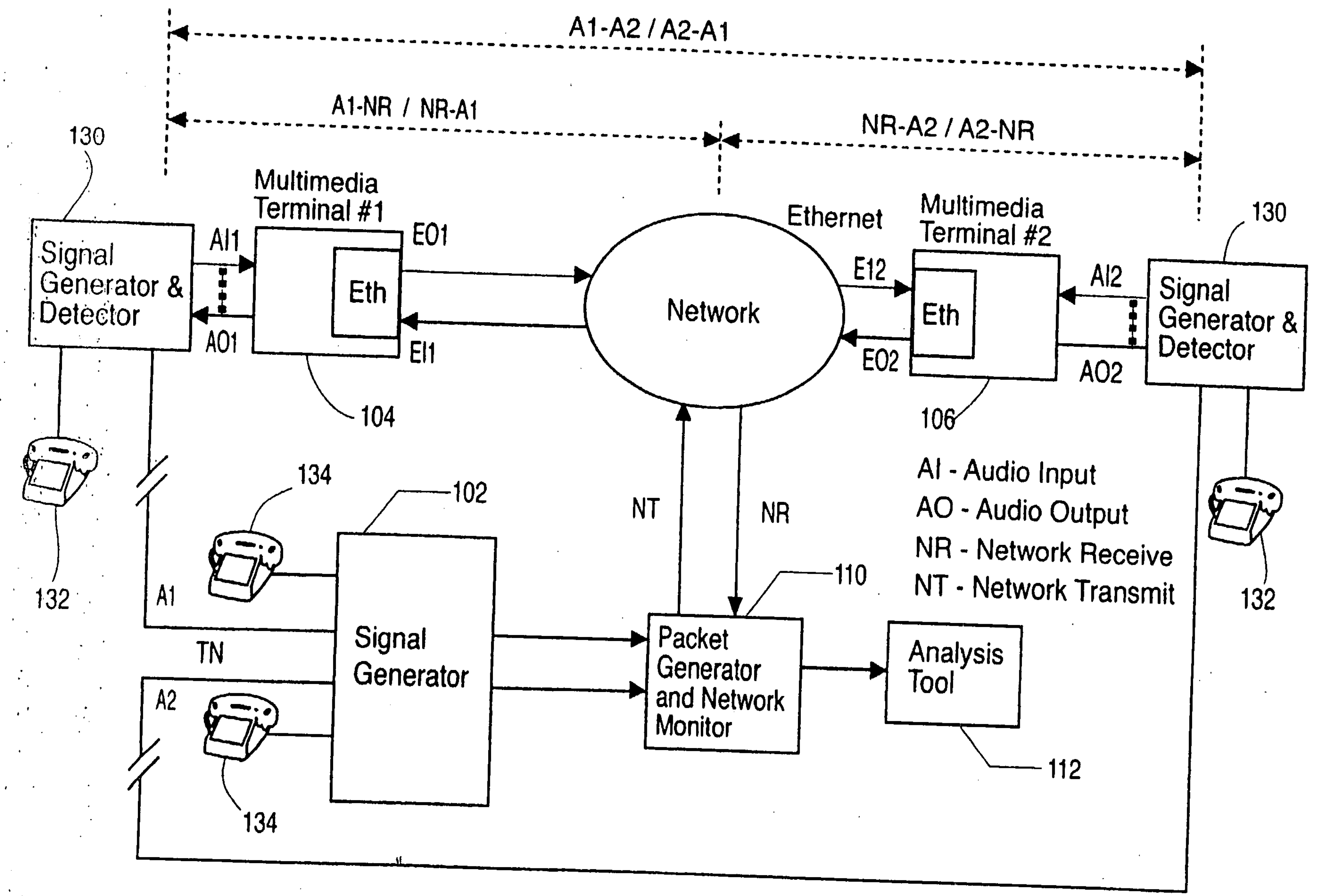

AI1-AO2
AI1-EO1
EO1-AO2
Multimedia Terminal #1
Multimedia Terminal #2
EO1
EI1
AI1
AO1
Network
E12
EO2
AI2
AO2
Pulse Generator
Trigger Circuit
Packet Generator and Network Monitor
Analysis Tool
AI - Audio Input
AO - Audio Output
EI - Network Input
EO - Network Output

Figure 5



A1-A2 / A2-A1
A1-NR / NR-A1
NR-A2 / A2-NR
130
Signal Generator & Detector
AI1
AO1
Multimedia Terminal #1
Eth
EO1
EI1
104
Network
Ethernet
E12
EO2
Multimedia Terminal #2
Eth
106
AI2
AO2
130
Signal Generator & Detector
132
134
102
Signal Generator
A1
TN
A2
134
NT
NR
110
Packet Generator and Network Monitor
Analysis Tool
112
AI - Audio Input
AO - Audio Output
NR - Network Receive
NT - Network Transmit
132

MARKER PACKET SYSTEM AND METHOD FOR MEASURING AUDIO NETWORK DELAYS

FIELD OF THE INVENTION

The present invention relates generally to the field of electronic multimedia communications and more specifically to a system and method for measuring audio signal and packet delays across a network.

BACKGROUND OF THE INVENTION

With the increased deployment of local area and wide area networks within corporations and organizations, new technologies have emerged to take advantage of such network deployments. In traditional networks, voice communications are facilitated by direct, circuit switched connections; however, such networks are resource intensive and costly. Therefore, new technology has arisen that provides for the transmission of voice communications over packet transport networks that were originally designed primarily for data transmission. The transmission of voice over packet transport networks introduces new challenges into the design of not only the packet transport network but also the multimedia equipment connected to such networks to facilitate the voice communications. Voice communications are particularly sensitive to time delays, therefore it is extremely important to have efficient multimedia equipment for receiving and generating audio signals, placing them on the network, as well as efficient packet transport network communications equipment and an efficient network topology. There is a need in the art for a system and method to measure the delays in multimedia equipment and across packet transport networks to aid in the design, configuration and development of efficient multimedia terminal equipment, as well as the design and development of efficient network equipment and typologies. In addition, there is a need in the art for a system and method for effective isolation and the removal of network bottlenecks in voice communications.

SUMMARY OF THE INVENTION

The present invention provides a system and method to measure the delays in multimedia equipment and across packet transport networks to aid in the design,

configuration and development of efficient multimedia terminal equipment and software, as well the design and development of efficient network equipment, software and typologies. The present invention also aids in the isolation of network bottlenecks.

The present invention generates test audio signals for testing the network. In addition, the present invention uses a marker packet transmitted on the network to mark the occurrence of input and output audio signals. The system and method locates and captures the audio signal that has been encoded into an audio stream packet on the network and calculates the delays across multimedia terminal devices as well as across the network. The present invention can measure the delay between an analog audio signal and the occurrence of the subsequent audio stream packet on the network containing the audio signal. Furthermore, the present invention can measure the delay between the occurrence of an audio stream packet on a network and the subsequent generation of the corresponding registered audio signal by the receiving equipment. In addition, the present invention can measure the end to end signal delay from audio signal creation at the start to audio signal generation by a receiver at the destination.

The present invention consists of a preferred and alternate embodiments that may be used to measure packetized voice delays at different points in a network that transmits information and data in packets.

According to one aspect of the present invention there is provided a system for measuring delays across equipment in a packet network comprising: (a) a signal generator for generating test audio signals and trigger signals; (b) one or more network terminals coupled to the network and the signal generator for receiving the test audio signals, coding and decoding audio stream packets, transmitting and receiving the audio stream packets on the network and regenerating audio signals; (c) a packet generator coupled to the network and the signal generator for generating marker packets on the network in response to the trigger signals; (d) a network monitor coupled to the network for capturing the marker packets and the audio stream packets and recording the packets to create captured data; and (e) a packet analyzer coupled to the network monitor for analyzing the captured data to generate and display measurements of delays.

According to another aspect of the present invention, there is provided a system for measuring delays across equipment in a packet network comprising: (a) a pulse generator for generating test audio signals; (b) one or more multimedia terminals

coupled to the network and the pulse generator for receiving the test audio signals, coding and decoding audio stream packets, transmitting and receiving the audio stream packets on the network and regenerating audio signals; (c) a trigger circuit coupled to the pulse generator and the multimedia terminals for receiving regenerated audio signals and test audio signals and generating trigger signals; (d) a packet generator coupled to the network and the trigger circuit for generating marker packets on the network in response to the trigger signals; (e) a network monitor coupled to the network for capturing the marker packets and the audio stream packets and recording time stamps to create captured data; and (f) a packet analyzer coupled to the network monitor for analyzing the captured data to generate and display measurements of delays.

According to a further aspect of the present invention there is provided a method for measuring delays across equipment in a packet network comprising the steps of: (a) generating first marker packet on the network contemporaneously with the generation of a test audio signal to a network terminal; (b) generating an audio stream packet by the network terminal on the network containing an encoded audio signal corresponding to the test audio signal; (c) capturing the first marker packet and the audio stream packet with associated time stamps corresponding to moment of capture; and (d) subtracting the associated time stamps corresponding to the first marker packet and the audio stream packet to determine raw delay across the network terminal.

In addition, according to a further aspect of the present invention, there is provided a method for measuring delays across equipment in a network comprising the steps of: (a) generating first marker packet on the network contemporaneously with the generation of a test audio signal to a first network terminal; (b) generating an audio stream packet by first network terminal on the network containing an encoded audio signal corresponding to the test audio signal; (c) regenerating the test audio signal on a second network terminal in response to receipt of the audio stream packet on the second network terminal; (d) generating a second marker packet on network contemporaneously with the regenerating of test audio signal on the second network terminal; (e) capturing the first marker packet and the second marker packet with associated time stamps corresponding to moment of capture; and (f) subtracting the associated time stamps corresponding to the first marker packet and the second marker packet to determine end to end delay across the network.

In addition, according to a further aspect of the present invention there is provided a method for measuring delays across equipment in a network comprising the steps of: (a) generating first marker packet on said network contemporaneously with the generation of a test audio signal to a first network terminal; (b) generating an audio stream packet by first network terminal on said network containing an encoded audio signal corresponding to said test audio signal; (c) regenerating said test audio signal on a second network terminal in response to receipt of said audio stream packet on said second network terminal; (d) generating a second marker packet on network contemporaneously with said regenerating of test audio signal; (e) capturing said first marker packet, said second marker packet and said audio stream packets with associated time stamps corresponding to moment of capture; (f) subtracting said associated time stamps corresponding to said first marker packet and said second marker packet to determine end to end delay across said network; and (g) subtracting the associated time stamps corresponding to the second marker packet and the audio stream packet to determine raw delay across the second network terminal.

BRIEF DESCRIPTION OF THE DRAWINGS

A detailed description of the preferred embodiments is provided hereinbelow, with reference to the following drawings in which:

Figure 1 is a diagrammatic representation of an environment illustrating components of the present invention;

Figure 2 is a diagrammatic representation of system delays that may be compensated for in the environment of Figure 1;

Figure 3 is a diagrammatic representation of packets on the network in the environment of Figure 1, with delay (Figure 3A) and without delay (Figure 3B);

Figure 4 is a diagrammatic representation of a first alternate embodiment of components of the present invention;

Figure 4A is a diagrammatic representation of a second alternate embodiment of components of the present invention; and

Figure 5 is a diagrammatic representation of a third alternate embodiment of components of the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Turning to Figure 1, a typical environment 100 of the present invention is shown. The measurement system and method of the present invention uses a signal generator 102 connected to one or more network devices for generating and receiving signals from the network devices. In a preferred embodiment, the network devices are network terminals that are sound enabled such that they can generate and receive audio signals. Preferably, they are multimedia terminals 104 and 106, which may be personal computers as are known in the art with graphical user interfaces, and may run operating systems such as Windows 95 or other well known operating systems. Multimedia terminals 104 and 106 may employ commercially available sound cards or devices to facilitate the audio function. The multimedia terminals 104 and 106 are connected to a packetized network 108 with commercially available network interface cards or network interfaces and in a preferred embodiment, multimedia terminals 104 and 106 utilize commercially available software and hardware employing the public standard ITU-T Recommendation G.71 1 (PCM) coding for digital representation of audio signals transmitted or received on network 108. Alternatively, network terminals may be dedicated multimedia terminal devices with operating functions coded in hardware or firmware. Network 108 may be any type of commercially available packetized network, and in the preferred embodiment, network 108 uses Ethernet protocols. Alternatively, network 108 may incorporate multiple network segments, bridges, routers, wide area network (WAN) and Internet connections. The invention may be adapted in a manner obvious to one skilled in the art to operate on networks using protocols other than Ethernet.

The inventive system and method also uses a component for marker packet generation connected to both signal generator 102 and network 108. It also employs a component for network monitoring connected to network 108. The packet generation component and the network monitoring component may be combined into a single device shown as packet generator and monitor 110 in Figure 1.

Furthermore, the inventive system and method uses a packet analyzer 112, which is preferably coupled to the network monitor, for performing the measurement and analysis functions. The analyzer 112 also may have a display for output of results and memory for storage of measurement results. The analyzer 112 may be embodied in

software running on a typical personal computer connected to the networking monitoring component by a serial or port or other appropriate means.

The basic operational principles of the preferred embodiment of the present invention are described below. A test audio signal is generated by signal generator 102 that is received by multimedia terminal 104. Multimedia terminal 104 digitally encodes the test audio signal to create a corresponding audio stream packet, and transmits the audio stream packet onto network 108 for reception by multimedia terminal 106. The audio stream packet is recognizable by the networking monitoring component and when multimedia terminal 106 receives it from the network, it decodes and converts the signal back to audio and regenerates the audio signal as output. The regenerated audio signal is sent to and received by signal generator 102. At various stages, marker packets are generated by the packet generator component triggered by the signal generator 102 to mark events, and network traffic is logged with time stamps and analyzed to determine the delays at various stages. The operation of the system can also be reversed, so that the signal generator 102 generates an audio signal for multimedia terminal 106 that is eventually regenerated in a similar manner, on multimedia terminal 104. Optionally, audio stream packets may travel in both directions at the same time.

The signal generator 102 is connected to the inputs and outputs of the multimedia terminals 104 and 106, preferably audio input 114 and audio output 118 of multimedia terminal 104 and audio input 116 and audio output 120 of multimedia terminal 106 for the generation and reception of audio signals. The signal generator 102 may be connected to the multimedia terminals using standard cables as are well known in the art. The signal generator 102 is also connected the packet generation component of packet generator and monitor 110 using standard cables. Signal generator 102 generates preferably analogue audio source signals which are input into audio inputs 114 and 116 of multimedia terminals 104 and 106 respectively, and also receives the regenerated audio signals from the outputs 118 and 120 of the respective multimedia terminals 104 and 106.

In the preferred embodiment, the signal generator 102 is adapted to generate a test audio signal to multimedia terminals 104 and 106 that is analogue in the form of a voltage and has pulse shape, which is convenient for detection. However, the invention is not restricted to the pulse shaped signal only. Other signal shapes are possible. This

test audio signal is used to initiate a delay measurement, and it is generated repeatedly at an adjustable rate by an adjustment in signal generator 102. Preferably, the period of repetition is chosen long enough to accommodate the largest possible end-to-end delays that are expected to occur, but short enough to make measurement convenient. No overlap of test audio signals in any particular direction should be allowed, such that the test audio signal originated by the signal generator 102 and sent through multimedia terminal 104 must be propagated through the network 108 and must appear on the multimedia terminal 106 audio output 120 before the next test audio signal is generated by signal generator 102. The voltage level should be adjusted to nominal audio levels to prevent damage to the equipment. In the preferred embodiment, the pulse width of the test audio signal is chosen to be 1ms, but larger pulse widths can also be used. Shorter pulses than 1ms are not preferred as they can be hard to detect. The test audio signals generated by the signal generator 102, as well as the regenerated audio signals from the far end (i.e. multimedia terminal 106), are used to initiate the generation of marker packets onto the network 108 by the packet generator component of packet generator and monitor 110.

The packet generation component of packet generator and monitor 110 is adapted and used to generate marker packets on network 108 that are unique and easily recognizable for capture from the network 108. The nature of the marker packets and the audio stream packets is that they can be recognized, captured and analyzed. Marker packets can be created, identified and recognized by choosing a specific size for the packet, or designating a specific pattern of characters within the packet generator component of packet generator and monitor 110. The network monitor component of packet generator and monitor 110 is adapted and configured in a manner to capture the network traffic containing the marker packets and the audio stream packets. With each packet captured, the network monitor component records a time stamp associated with the packet. The network monitor component of the packet generator and monitor 110 captures the network traffic, and may produce a hexadecimal trace file that may be analyzed off-line by the analysis tool 112. Alternatively, the network monitor component may funnel the captured traffic and time stamps directly to the analysis tool 112.

When the signal generator 102 generates a test audio signal that is input on

multimedia terminal 104, a trigger signal, which may be the same test audio signal, is simultaneously sent to the packet generator component of packet generator and monitor 110.

Marker packets are thus contemporaneously generated and introduced onto the network 108 by the packet generator component of the packet generator and network monitor 110 on the rising edge of source test audio signal created by signal generator 102.

Furthermore, signal generator 102 also acts as a relay of output signals generated by multimedia terminals 106 and 104. When an output signal is regenerated by either of multimedia terminals 104 or 106, a similar trigger signal as described above is created by signal generator 102 on receipt of the rising edge of the regenerated audio output signal and is sent to the packet generator component of the packet generator and monitor 110. Since some multimedia systems can cause inversion of audio signals, the signal generator 102 may be programmed to invert the regenerated audio output signal. Trigger signals sent from signal generator 102 to packet generator and monitor 110 may be sent on different signal lines depending upon whether they are input or output signals and whether they are from multimedia terminal 104 or 106 to allow the packet generator to distinguish the signals and generate an identifiable corresponding marker packet. Alternatively, the trigger signals could be distinguished using some other means, such as voltage, etc.

In the preferred embodiment, audio delays can be measured between several reference points in the environment 100. For notational purposes, the reference points are referred to as follows: AI1 represents the audio input 114 of multimedia terminal 104. AOl represents the audio output 118 of multimedia terminal 104. EI1 and EO1 represents the network input and output respectively of multimedia terminal 104. In a similar manner: AI2, AO2, EI2 and EO2 represent the corresponding inputs and outputs on multimedia terminal 106. Using this notation, AIl-EO1 is the audio delay from the analog audio input of the multimedia terminal 104 to the network output of multimedia terminal 104. Analogously, the other audio delays that can be measured are EI1-AO1, EO1-AO2, AI2-EI1, and the end-to-end delays AI1-AO2, AI2-AO1.

In another embodiment of the present invention, to measure delays in both directions, i.e. transmit and receive direction, through the multimedia terminals 104 and

106 at the same time, two different marker packets are injected into the network, one type of marker packet for each direction. As mentioned above, marker packets can be made unique by choosing a specific size for the network packet or by a specific pattern of characters and by configuring the network monitoring tool and analysis tool to recognize and analyze such packets accordingly.

Returning to Figure 1, as mentioned above, signal generator 102 sends a trigger signal unique to the event to the packet generator component of packet generator and monitor 110 to generate a marker packet. The marker packet can be used to identify a particular event such as the origination of a test audio signal or receipt of a regenerated audio signal. The network monitoring features of packet generator and monitor 110 is adapted and configured to recognize packets, and ensures that all network packets of interest and particularly marker packets and audio stream packets are captured and time-stamped.

After the desired number of measurements are taken and captured by the network monitor component, capturing is stopped, and optionally a hexadecimal trace file is produced.

The time stamp resolution, the delays of propagating signals across the packet generation component, and the delay of sending the marker packets will affect the measurement accuracy.

Specifically, the signal generator 102 and the packet generator and network monitor 110 introduce additional delays to signals traveling through these components. The signal generator 102 transforms an analog signal to a digital trigger signal with a certain delay, and monitor 110 sends a packet to the network with a certain delay after a trigger signal is received. These delays may not always be constant but the accuracy of delay measurements can be improved by taking into account these delays. Figure 2 illustrates the delays that may be compensated.

The packet generation delay affects only the packet delay. This delay is added to the non-compensated raw packet delay to get a more accurate packet delay. For the end-to-end delay calculations, compensation is not necessary because both the initial marker packet and the final one are delayed by the same amount. The end-to-end delay is additionally affected by the signal generator delay, which is subtracted from the non-compensated end-to-end delay. The compensation is carried out by the analysis tool

112. The packet generator delay is largely dependent on the computing platform used for that application. On the other hand, the signal generator delay is more constant and it depends on the electronic circuits for audio signal detection and signal shaping.

The analysis tool 112 is configured and adapted to recognize marker packets and audio stream packets and calculates the delay values as well as provide statistical analysis of the results. It may optionally provide off-line processing of the hexadecimal trace file captured by the network monitor. In a preferred embodiment, analysis tool 112 may be embodied in software on a typical personal computer connected by serial port, or other well known means, to the network monitor. Optionally, the hexadecimal trace file may be stored and later communicated by other means, such as floppy disk memory. The analysis tool 112 performs its subtraction measurement by using the difference in time between the time stamp on the marker packets and the time stamp on the corresponding audio stream packets. In the operation of the preferred embodiment, network packets of two types are generated to measure the delay AI1-EO1: (a) a marker packet on the generation of the test audio signal; and (b) a corresponding audio stream packet of the test audio signal. The time differential of these two packets can be used to calculate the delay through the multimedia terminal 104. When the audio stream packet is received by multimedia terminal 106, the regenerated test signal is produced. A third network packet, i.e. a marker packet, is generated when the audio is regenerated by multimedia terminal 106 and can be used to determine the network delay and multimedia terminal 106 delay. Using the timing information from the audio stream packet and the marker packet from the regenerated output audio signal from multimedia terminal 106, the delay EO1-AO2 can be calculated. Furthermore, using the two marker packets, the delay AI1-A02 can also be calculated. The procedure can also be operated in reverse with the signal generator 102 using a test audio signal to audio port 120 of multimedia terminal 106. In a manner obviously similar to that described above, the delays A12-EIl, EI1 to AO1 can be calculated. Similar packets to the procedure above are produced in reverse.

Audio stream packets can be identified on the network in the following manner. Each packet of the captured network traffic is examined, whether it is a marker packet or a packet containing the encoded audio source signal. Depending on which encoding law was used in the system, A-law or u-law (both of which are well known), and

depending on the input voltage level, the input pulse will be coded by a certain constant value for the duration of the pulse. If the G.711 standard is employed, then there are 8000 samples per second, and the encoding of the samples consists of 8 binary digits per sample, and an input pulse of 1ms duration will be encoded by 8 bytes in the payload of the audio stream packet. Longer input pulses would be coded proportionally by more bytes. Because variations of the input signal level can occur, a searching algorithm can be employed in the analysis tool which allows for a variation of byte values between two extreme values, and looks for a match of a predefined number of bytes that are consecutive in the packet payload, at certain offset from the beginning of the packet frame. The offset in the Analysis Tool 112 can be set to a selected value. Using as an example of the usage of RTP protocol for sequencing the audio packets, as the H.323 standard suggests, the offset for searching for the encoded pulse in the packet would be 54 bytes. This offset includes the MAC, IP, UDP and the RTP header.

After the successful search for the encoded audio pulse, the raw audio signal to packet delay is calculated by subtracting the time stamp of the last marker packet from the time stamp of the audio stream packet that contains the encoded audio pulse of the test audio signal; this is the raw delay AI1-EO1. The raw voice packet to regenerated audio delay (i.e., the raw delay EO1-AO2) is calculated by subtracting the time stamp of the packet that contains the encoded pulse from the marker packet time stamp that follows the former packet.

While the time stamps on the marker packets and the audio stream packets can be used to calculate the raw delay with a certain level of accuracy, further improved measurements can be performed by locating the position of the digitally encoded test signal in the audio stream packets as described in further detail with respect to Figure 2.

Consider an application wherein voice packet streaming occurs every 20 ms, i.e. every 20 ms an Ethernet packet is sent on the network. The G.711 standard specifies 8000 samples per second, and sample encoding by 8 bits. The 20 ms audio information is encoded by 160 samples, i.e. 160 bytes (0.020*8000). The RTP packet payload is therefore 160 bytes per packet. The total length of an Ethernet voice streaming packet is thus 218 bytes: 54 bytes header (MAC, IP, UDP and RTP header), 160 bytes payload bytes and 4 bytes FCS (Ethernet Frame Check Sequence). The Ethernet packet starts with a preamble followed by the MAC header and all other (IP, UDP, RTP) headers,

followed by the RTP payload data, and finally the FCS. The network monitor 110 takes time stamp information after the packet as a whole (after receiving the last byte of FCS) has been received. The duration of the packet on a 10 Mbit/s network is 0.174ms (i.e. 218*8*0.0000001). The audio input signal (1ms in duration) is encoded by 8 bytes. The packet abstraction of Figure 3A represents the 20 ms audio information, which ends when a voice packet (i.e. an Ethernet packet that contains the coded audio test (voice) signals) is generated (the time stamp is taken by the network monitor 11), and it starts 20 ms earlier. The RPT payload bytes at the beginning of the payload (left-hand side) of Figure 3A represent audio samples that occurred earlier in the past; the last byte of the payload (right-hand side) of Figure 3A represents the most recent audio sample. Because of processing overhead, the multimedia terminal 104 introduces a delay, which results in shifting the audio input samples (and all the samples) toward the end of the RTP payload (the audio input does not appear aligned with the encoded audio input in the payload). Shifting the audio input samples (bytes) in the payload causes delays in regeneration of the audio output signal.

The audio-to-packet delay is calculated in the following manner. Firstly, the analysis tool 112 calculates the raw audio-to-packet delay as a time difference between the time stamp of the packet containing the encoded audio signal and the time stamp of the marker packet preceding that packet (e.g. 17ms). Next, the analysis tool 112 detects the position of the beginning of the encoded audio input signal from the end of the payload, which can be e.g. 48 (maximum 160). Each byte represents a time duration of 0.125ms (1/8000s), i.e. 1ms audio signal is encoded by 8 bytes. 48 bytes are converted to a time value of 6ms, i.e. 48*0.125ms. The packet delay introduced by the multimedia terminal 104 (106) in the transmit direction is calculated by subtracting this time value from the raw packet delay, i.e. 17ms - 6ms = 11ms.

In the receive direction, the analysis tool 112 first calculates the raw packet-to-audio delay by subtracting the time stamp of the marker packet that follows the Ethernet packet containing the audio signal and the time stamp of the Ethernet packet containing the audio signal (e.g. 40 ms). Next, the analysis tool detects the position of the beginning of the encoded audio signal from the beginning of the RTP payload, which can be e.g. 56 (maximum 160). 56 sample intervals transforms into a time value of 7ms (56*0.125ms). The packet-to-audio delay introduced by the multimedia terminal 104

(106) in the receive direction is calculated by subtracting this time value from the raw packet-to-audio delay, i.e. 40ms - 7ms = 33ms.

In order to better illustrate the extent of the introduced delays, Figure 3B shows a scenario wherein the multimedia terminals 104 and 106 introduce no additional delays. Even if the multimedia terminals do not introduce delays, end-to-end delay still exists as a result of buffering (20 ms) and delays introduced by the network.

Figure 4 shows an alternate embodiment of the present invention. This alternate embodiment is somewhat simpler and takes measurements only in one direction at a time across the multimedia terminals 104 and 106. The role of the signal generator 102 of Figure 1 is divided into a pulse generator 302 and a trigger circuit 304 in this alternate embodiment which simplifies the components of the inventive system. To get measurements in both transmit and receive directions across a multimedia terminal, the measurement has to be carried out twice; once feeding the audio input source signal from pulse generator 302 into audio input 114 of the multimedia terminal 104, and for the second measurement feeding the input signal into audio input of the multimedia terminal 106. The pulse generator 302 also sends the test audio signal to the trigger circuit 304 which in turn issues a trigger signal to packet generator and network monitor 110 to issue a marker packet. In a simpler manner as described with respect to Figure 1, trigger circuit 304 also issues a marker packet on receipt of a regenerated audio signal from multimedia terminal 106. In other respects, this alternate embodiment operates in a manner obviously similar to that previously described with respect to Figure 1. Drawbacks of this simpler variant are that the measurement takes longer time to complete, and that no correlation of delays in transmit and receive direction across the multimedia terminals 104 and 106 at the same time can be given.

A second alternative embodiment is depicted in Figure 4A wherein a loopback circuit is provided at the remote multimedia terminal, whereby the regenerated audio test signal is looped back through a loopback circuit electronically, acoustically or using a loopback feature of the multimedia terminal. Preferably this embodiment can be used in the field, where a loopback circuit at points AO2-AI2 may be installed, possibly even by a non-technician.

The benefit of this embodiment is that without having to provide an extra physical connection to the remote multimedia terminal, delays across the local

multimedia terminal can be determined as well as the loopback delay, which can be used to estimate the end-to-end delay. The loopback delay consists of the end-to-end delay from the local multimedia terminal to the remote multimedia terminal and additionally the end-to-end delay from the remote multimedia terminal to the local multimedia terminal where the measurement equipment is located.

The embodiment of Figure 1 assumes that the multimedia terminals 104 and 106 are located in close proximity to be able to connect both terminals to the signal generators 102. It is also possible to apply the principles of the invention remotely with minor modifications to the embodiment of Figure 1. Figure 2 is similar to Figure 1 with the exception of additional components in the form of signal generator and detector 130 and telephones 132 and 134 connected over a telephone network (TN).

Multimedia terminals 104 and 106 are accessed remotely, using manual or automated dialing and the phone calls placed thereby are answered either manually or using auto-answer capability. The signal generator and detector 130, as well as signal generator 102, are provided telephone line access capability. The dotted line between reference points AI1 and AO1 represent a loopback capability. This loopback capability is necessary to measure the delay over the telephone lines from the signal generator 102 to the remote signal generator and detector 130 at the multimedia terminals 104 and 106 and back to signal generator 102. The delay from A1 to AI1 (or A2 to AI2) is one-half of the measured loopback delay, assuming a circuit switched connection. The loopback measurements can be carried out using the arrangement shown in Figure 5.

The purpose of the signal generator and detector 130 is to detect the transmitted audio signal from signal generation 102 and generate a pulse signal to AI1 (A12) analog input of the multimedia terminal 104. The packet generator and network monitor 110 is connected at a desired access point to the network. The packet delay measurements also include the delay (A1-AI1/AO1-A1 and A2-AI2/AO2-A2) over the telephone lines.

Although the invention has been described in terms of the preferred and several alternate embodiments described herein, those skilled in the art will appreciate other embodiments and modifications which can be made without departing from the spirit and scope of the teachings of the invention. All such modifications are intended to be included within the scope of the claims appended hereto.

Claims:

1. A system for measuring delays across equipment in a packet network comprising:

(a) a signal generator for generating test audio signals and trigger signals;

(b) one or more network terminals coupled to said network and said signal generator for receiving said test audio signals, coding and decoding audio stream packets, transmitting and receiving said audio stream packets on said network and regenerating audio signals;

(c) a packet generator coupled to said network and said signal generator for generating marker packets on said network in response to said trigger signals.

(d) a network monitor coupled to said network for capturing said marker packets, said audio stream packets and recording said packets to create captured data; and

(e) a packet analyzer coupled to said network monitor for analyzing said captured data to generate and display measurements of delay.

2. The system of claim 1 wherein said network uses Ethernet protocols.

3. The system of claim 1 wherein said packet generator and said network monitor are combined into a single device.

4. The system of claim 1 wherein said test audio signal is a pulse of approximately 1 millisecond duration.

5. A system for measuring delays across equipment in a packet network comprising:

(a) a pulse generator for generating test audio signals;

(b) one or more multimedia terminals coupled to said network and said pulse generator for receiving said test audio signals, coding and

decoding audio stream packets, transmitting and receiving said audio stream packets on said network and creating regenerated audio signals;

(c) a trigger circuit coupled to said pulse generator and said multimedia terminals for receiving regenerated audio signals and test audio signals and generating trigger signals;

(d) a packet generator coupled to said network and said trigger circuit for generating marker packets on said network in response to said trigger signals;

(e) a network monitor coupled to said network for capturing said marker packets and said audio stream packets and recording time stamps to create captured data; and

(f) a packet analyzer coupled to said network monitor for analyzing said captured data to generate and display measurements of delays.

6. The system of claim 5, further comprising a loopback circuit for feeding back said regenerated audio signals from an output to an input of said one or more multimedia terminals.

7. A method for measuring delays across equipment in a packet network comprising the steps of:

(a) generating a first marker packet on said network contemporaneously with the generation of a test audio signal to a network terminal;

(b) generating an audio stream packet by said network terminal on said network containing an encoded audio signal corresponding to said test audio signal;

(c) capturing said first marker packet and said audio stream packet with associated time stamps corresponding to moment of capture; and

(d) subtracting said associated time stamps corresponding to said first marker packet and said audio stream packet to determine raw delay across said network terminal.

8. The method of claim 7 comprising the further steps of:

(a) analyzing said audio packet to determine time offset of said encoded audio signal from end of said audio stream packet; and

(b) subtracting said time offset from said raw delay to obtain net delay across said network terminal.

9. A method for measuring delays across equipment in a network comprising the steps of:

(a) generating a first marker packet on said network contemporaneously with the generation of a test audio signal to a first network terminal;

(b) generating an audio stream packet by first network terminal on said network containing an encoded audio signal corresponding to said test audio signal;

(c) regenerating said test audio signal on a second network terminal in response to receipt of said audio stream packet on said second network terminal;

(d) generating a second marker packet on network contemporaneously with said regenerating of test audio signal on said second network terminal;

(e) capturing said first marker packet and said second marker packet with associated time stamps corresponding to moment of capture; and

(f) subtracting said associated time stamps corresponding to said first marker packet and said second marker packet to determine end-to-end delay across said network.

10. A method for measuring delays across equipment in a network comprising the steps of:

(a) generating first marker packet on said network contemporaneously with the generation of a test audio signal to a first network terminal;

(b) generating an audio stream packet by first network terminal on said network containing an encoded audio signal corresponding to said

test audio signal;

(c) regenerating said test audio signal on a second network terminal in response to receipt of said audio stream packet on said second network terminal;

(d) generating a second marker packet on network contemporaneously with said regenerating of test audio signal;

(e) capturing said first marker packet, said second marker packet and said audio stream packets with associated time stamps corresponding to moment of capture;

(f) subtracting said associated time stamps corresponding to said first marker packet and said second marker packet to determine end-to-end delay across said network; and

(g) subtracting said associated time stamps corresponding to said second marker packet and said audio stream packet to determine raw delay across said second network terminal.

11. The method of claim 10 comprising the further steps of:

(a) analyzing said audio stream packet to determine time offset of said encoded audio signal from start of said audio stream packet; and

(b) subtracting said time offset from said raw delay to obtain net delay across said second network terminal.

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Claims searched: All
Examiner: Gareth Griffiths
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Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.R): H4K (KTL), H4P (PEUL, PEUM, PEUX, PPD, PPS)

Int Cl (Ed.7): H04L 12/26, 12/56

Other: Online Databases: WPI, EPODOC, JAPIO

Documents considered to be relevant:

Category	Identity of document and relevant passage		Relevant to claims
A	GB2261799 A	(DOWTY)	
A	EP0234860 A2	(AT&T)	
A	US5793976	(CHEN)	

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